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H3U UGX
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EP 0203726 A1 US 4633482 A

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(54) Recovering a time-varying signal using multiple adaptive filtering algorithms

(57) For recovering symbols in a data packet (101), Fig. 1, (not shown), transmitted to a receiver (201), Fig. 2, (not shown), from a remote signal source (202) in a time-varying channel using multiple adaptive filtering algorithms, there is used an adaptive filter 203 responsive to a recursive least squares (RLS) filtering algorithm followed by a least-mean squares (LMS) filtering algorithm. The RLS filtering algorithm is used first for its fast training and fast recovery characteristics. The LMS filtering algorithm is used second for its low complexity and high stability characteristics. The switch between the RLS and the LMS filtering algorithms may be at a fixed point in time to ensure minimal filter complexity or may be adaptive responsive to the quality of an error signal (e_k) processed in a coefficient determinator (307 or 309) to ensure maximum bit error rate performance and stability.

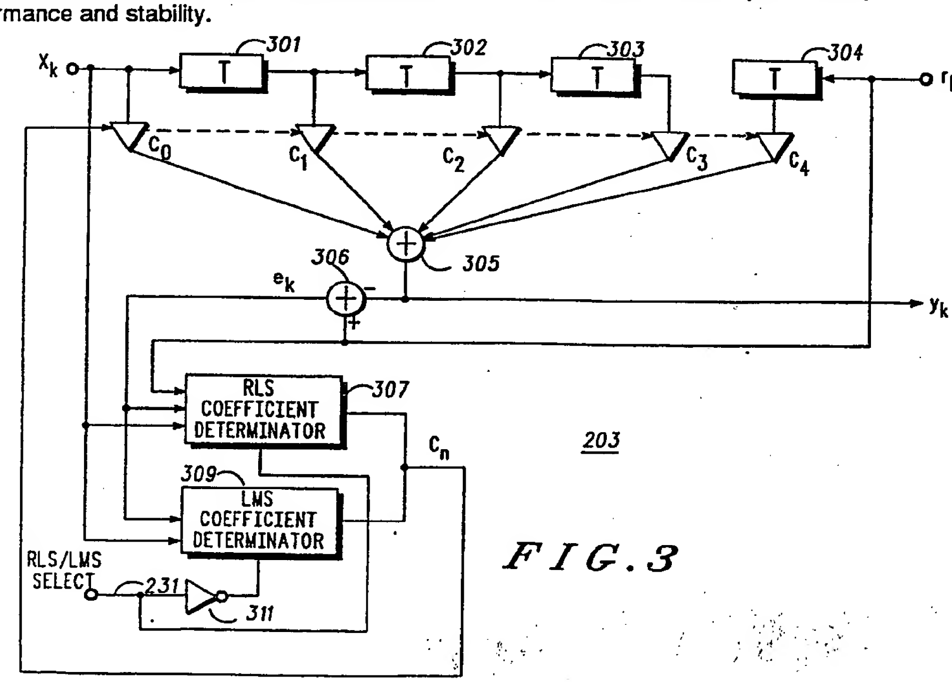

FIG. 3

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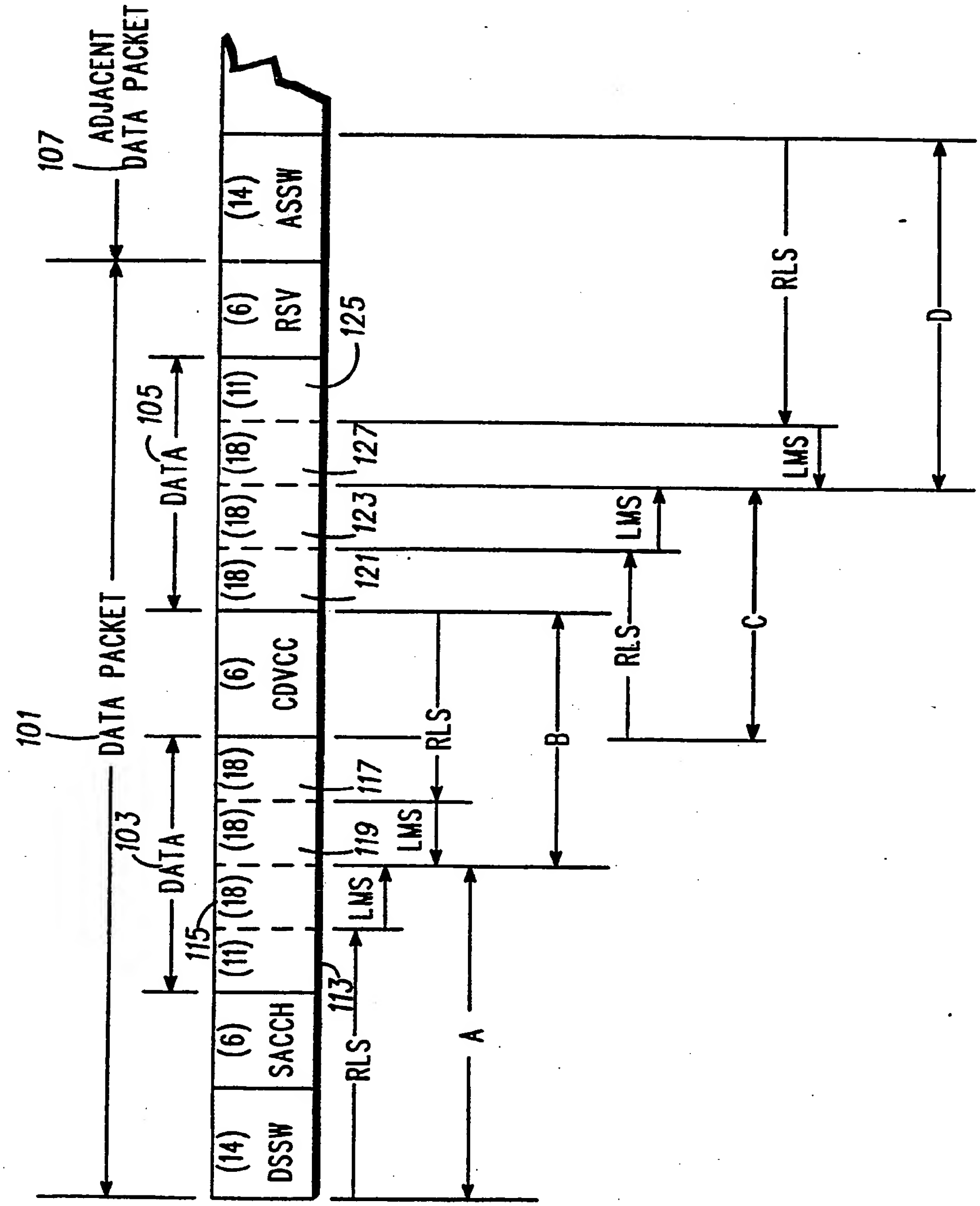
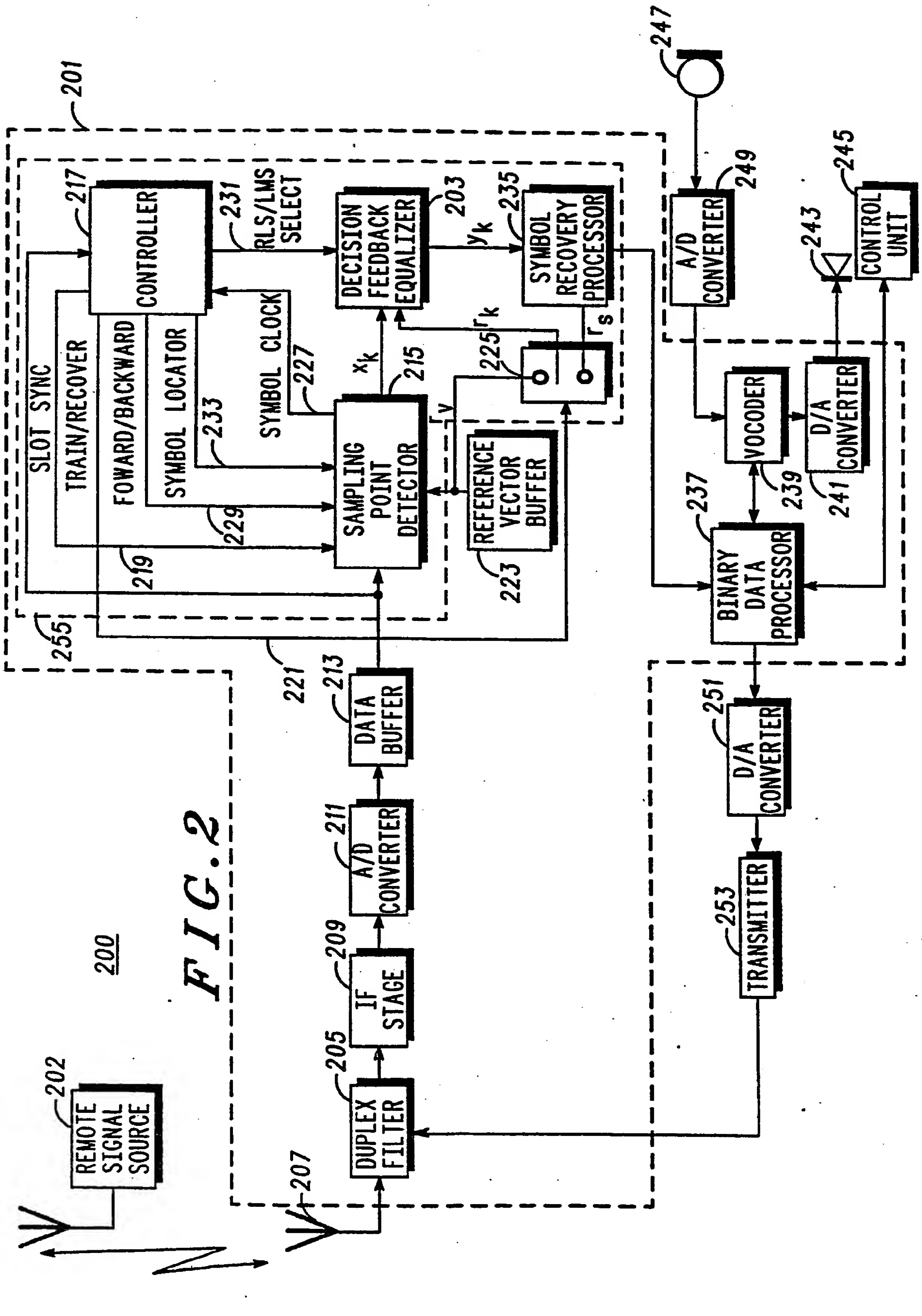
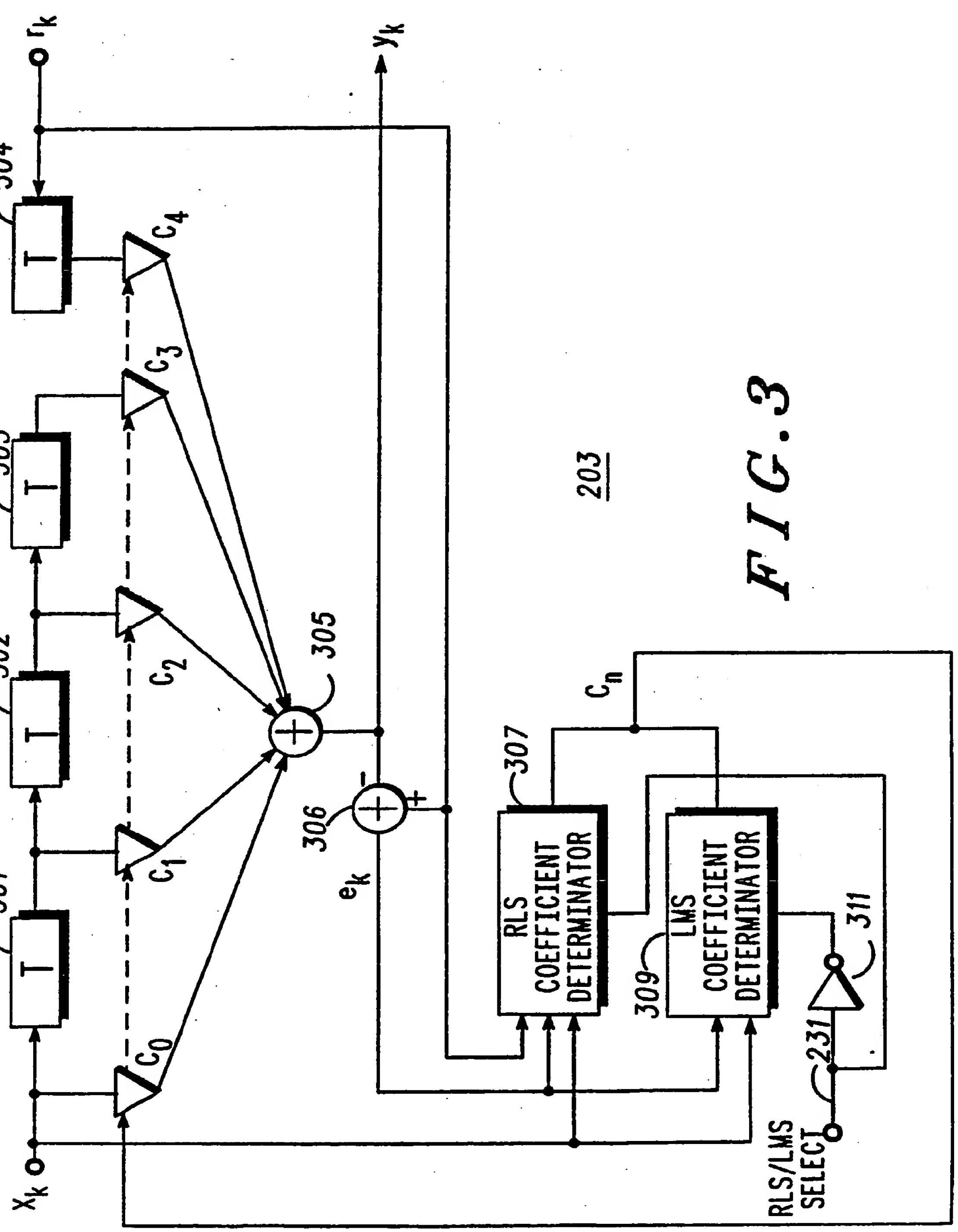

107
ADJACENT DATA PACKET
101
DATA PACKET
103
DATA
105
DATA
(14) DSSW
(6) SACCH
(11)
(18)
(18)
(18)
(6) CDVCC
(18)
(18)
(18)
(11)
(6) RSV
(14) ASSW
113
115
119
117
121
123
127
125
RLS
LMS
LMS
RLS
RLS
LMS
LMS
RLS
A
B
C
D

FIG.1



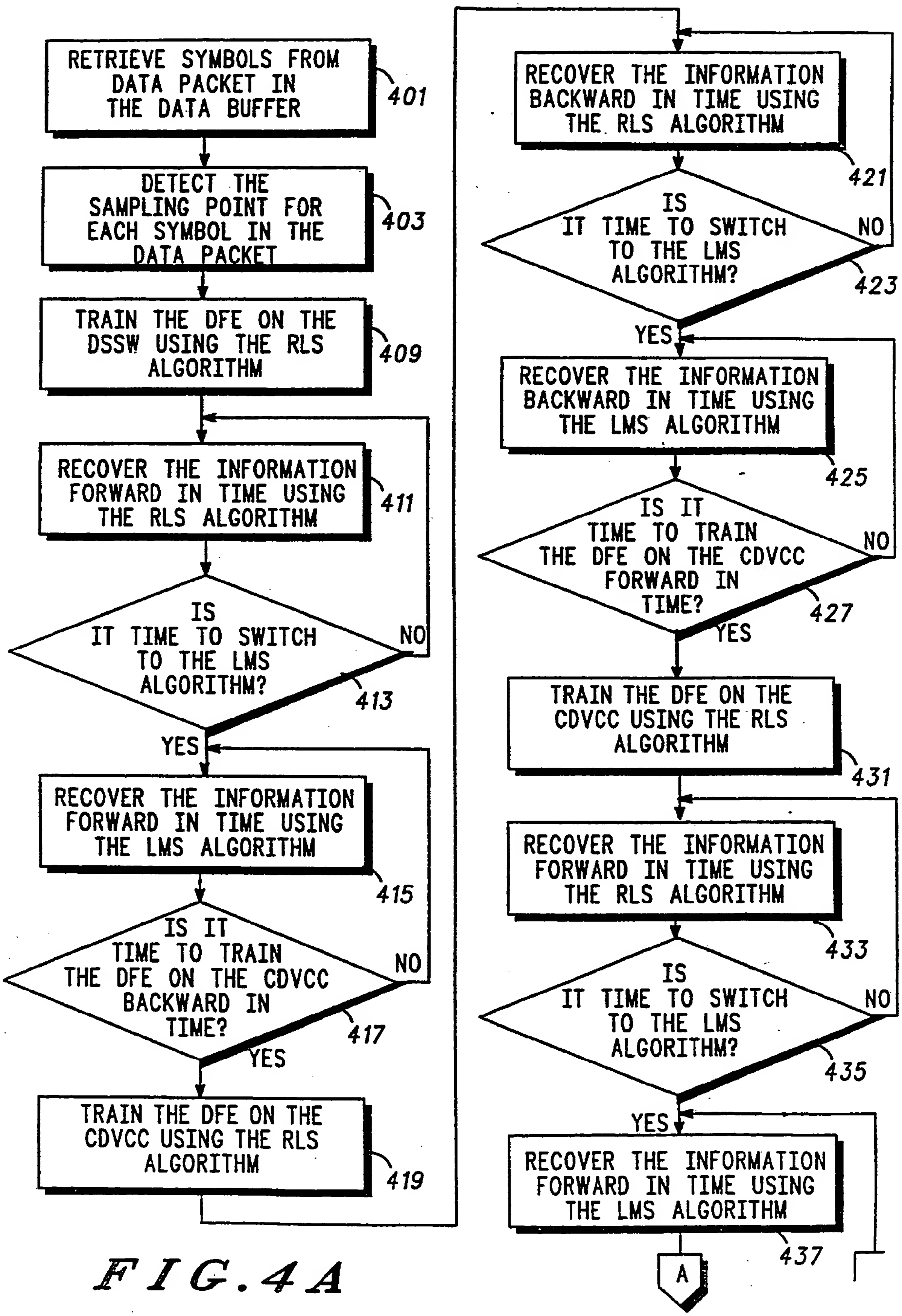
200
201
202
REMOTE SIGNAL SOURCE
207
205
DUPLEX FILTER
209
IF STAGE
211
A/D CONVERTER
213
DATA BUFFER
221
255
217
CONTROLLER
231
RLS/LMS SELECT
203
DECISION FEEDBACK EQUALIZER
y_k
235
SYMBOL RECOVERY PROCESSOR
r_k
r_s
225
r_v
SLOT SYNC
TRAIN/RECOVER
FOWARD/BACKWARD
SYMBOL LOCATOR
233
SYMBOL CLOCK
227
x_k
215
SAMPLING POINT DETECTOR
229
219
REFERENCE VECTOR BUFFER
223
247
249
A/D CONVERTER
243
245
CONTROL UNIT
VOCODER
239
D/A CONVERTER
241
237
BINARY DATA PROCESSOR
251
D/A CONVERTER
253
TRANSMITTER

FIG. 2



r_k
y_k
304
T
c_4
303
T
c_3
302
T
c_2
305
301
T
c_1
c_0
x_k
306
e_k
c_n
307
RLS COEFFICIENT DETERMINATOR
309
LMS COEFFICIENT DETERMINATOR
311
231
RLS/LMS SELECT
203

FIG.3



RETRIEVE SYMBOLS FROM DATA PACKET IN THE DATA BUFFER
401
DETECT THE SAMPLING POINT FOR EACH SYMBOL IN THE DATA PACKET
403
TRAIN THE DFE ON THE DSSW USING THE RLS ALGORITHM
409
RECOVER THE INFORMATION FORWARD IN TIME USING THE RLS ALGORITHM
411
IS IT TIME TO SWITCH TO THE LMS ALGORITHM?
NO
413
YES
RECOVER THE INFORMATION FORWARD IN TIME USING THE LMS ALGORITHM
415
IS IT TIME TO TRAIN THE DFE ON THE CDVCC BACKWARD IN TIME?
NO
417
YES
TRAIN THE DFE ON THE CDVCC USING THE RLS ALGORITHM
419
RECOVER THE INFORMATION BACKWARD IN TIME USING THE RLS ALGORITHM
421
IS IT TIME TO SWITCH TO THE LMS ALGORITHM?
NO
423
YES
RECOVER THE INFORMATION BACKWARD IN TIME USING THE LMS ALGORITHM
425
IS IT TIME TO TRAIN THE DFE ON THE CDVCC FORWARD IN TIME?
NO
427
YES
TRAIN THE DFE ON THE CDVCC USING THE RLS ALGORITHM
431
RECOVER THE INFORMATION FORWARD IN TIME USING THE RLS ALGORITHM
433
IS IT TIME TO SWITCH TO THE LMS ALGORITHM?
NO
435
YES
RECOVER THE INFORMATION FORWARD IN TIME USING THE LMS ALGORITHM
437
A

FIG. 4A

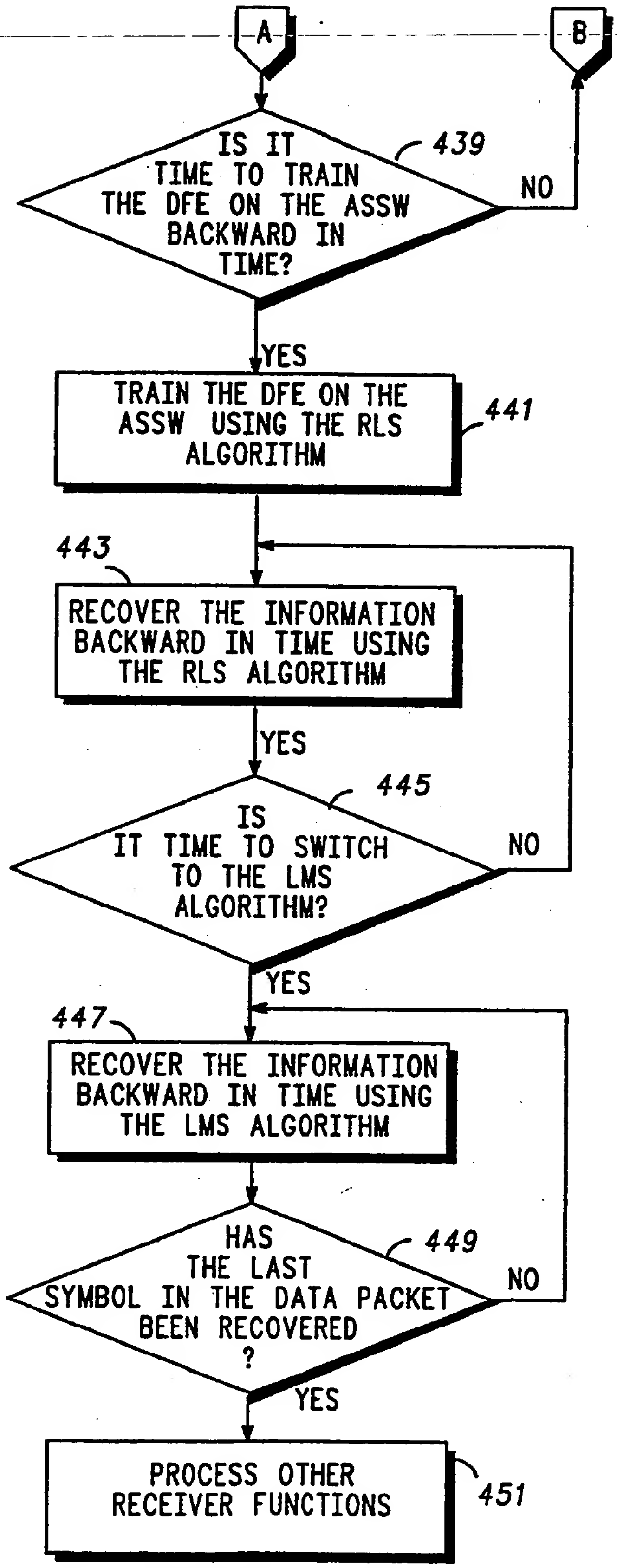

A
B
IS IT TIME TO TRAIN THE DFE ON THE ASSW BACKWARD IN TIME?
439
NO
YES
TRAIN THE DFE ON THE ASSW USING THE RLS ALGORITHM
441
443
RECOVER THE INFORMATION BACKWARD IN TIME USING THE RLS ALGORITHM
YES
445
IS IT TIME TO SWITCH TO THE LMS ALGORITHM?
NO
YES
447
RECOVER THE INFORMATION BACKWARD IN TIME USING THE LMS ALGORITHM
449
HAS THE LAST SYMBOL IN THE DATA PACKET BEEN RECOVERED ?
NO
YES
PROCESS OTHER RECEIVER FUNCTIONS
451

FIG.4B

2260068

Apparatus and Method for Adaptively Filtering a Time-Varying Signal Using Multiple Filtering Algorithms

Field of the Invention

The present invention relates generally to information recovery, and more particularly to an apparatus and method for adaptively filtering a time-varying signal using multiple filtering algorithms.

Background of the Invention

The rapid expansion of the number of cellular radio telephones coupled with the desire to provide additional services has prompted the development of a digital standard. The standard suggests an increase in system capacity over the previous analog system through the use of digital modulation and speech coding techniques. The standard for the cellular system is described in detail in Electronic Industries Association, Project Number 2398, January 1991, IS-54 (Revision A), entitled Dual-Mode Mobile Station -- Base Station Compatibility Standard. The standard describes in §1.2 a time division multiple access (TDMA) channel 40 milliseconds long divided into six equally sized data packets 6.66 milliseconds long. A data packet is a burst of information characterized by sequentially encoded consecutive pairs of bits, commonly known as symbols.

The standard describes in § 2.1.3.3.1 a linear modulation technique known as $\pi/4$ shifted, differentially encoded quadrature phase shift keying ($\pi/4$ DQPSK). The symbols are transmitted into one of four phase angles ($\pm\pi/4$, $\pm 3\pi/4$) using differential quadrature component signals producing an eight point phase constellation. The symbols are represented by a normalized magnitude vector and a phase angle. The symbols are transmitted as changes in phase rather than absolute phases.

Signal propagation in the radio frequency band, such as the 800 MHz band for cellular radiotelephones, is generally characterized by two types of channel-induced distortion: time dispersion distortion and multipath distortion. These types of distortion are caused by a rapid rate of change of the received data packet's

amplitude over time and are predominantly affected by the frequency of the signal, how rapidly the receiver is moving through its environment and large objects in the vicinity of the receiver. When the amplitude over a portion of the data packet approaches a null, the symbols can be corrupted by noise present in the channel that alters the state of the symbol causing the receiver to detect wrong information.

Time dispersion distortion is usually found in an environment where a large reflecting source, such as a mountain or a tall building, is present. A receiver operating in this environment receives the data packet from a fixed source transmitter and a delayed data packet from the reflecting source. The time delay between the reception of the two data packets results in time dispersion distortion.

Multipath distortion is characterized by many components of the same data packet having different energy levels reaching the receiver at the same time. As a result, the amplitude and phase of a data packet varies over time. This variance is referred to as "Rayleigh fading" of the data packet.

The present challenge is to recover received symbols in the data packet that were transmitted in the presence of the channel-induced distortion. Adaptive filtering for channel equalization is a widespread technique for removing the channel-induced distortion. Adaptive filtering may also be used in a wide variety of applications ranging from high resolution spectrum estimation and line enhancement, speech and biomedical signal processing, adaptive differential encoding and adaptive deconvolution to interference surpression and echo cancellation. Two data sequences are available to an adaptive filter, a desired response and an input. The

input sequence is applied to a linear filter with finite impulse response. The filter response is adapted so as to make its output sequence as close to as possible in some sense to the desired response sequence. A large number of equalizer adjustment algorithms are conceivable, depending on the system requirements.

One type of adaptive filtering algorithm is a recursive least squares (RLS) filtering algorithm. A description of the RLS filtering algorithm may be found in the IEEE (Institute of Electrical and Electronic Engineers) Transactions on Signal Processing, Volume 39, No. 1, January 1991, entitled "Numerically Stable Fast Transversal Filters for Recursive Least Squares Adaptive Filtering", presented by Dirk T.M. Slock and Thomas Kalith. The RLS filtering algorithm has fast training and recovery characteristics. These characteristics are important for a receiver to operate efficiently in a channel having time dispersion distortion. However, the RLS filtering algorithm is a complex calculation and tends to be unstable in a time-varying channel. The arithmetic operations of the RLS filtering algorithm require a high dynamic range that is very difficult to get in a fixed point digital signal processor and thus the RLS filtering algorithm becomes unstable. The complexity increases the receiver's processing time.

Another type of adaptive filtering algorithm is a least mean squares (LMS) filtering algorithm. A description of the LMS filtering algorithm may be found in The Bell System Technical Journal, October 1981, pages 1905-1925, entitled "Least-Squares Algorithms for Adaptive Equalizers" by M.S. Mueller. The LMS filtering algorithm has simple and very stable characteristics. These characteristics are important

for reducing the receiver's processing time and providing reliable data recovery. However, the LMS filtering algorithm has slow training and recovery characteristics that do not operate efficiently in a time-varying channel. The LMS filtering algorithm does not quickly converge for a rapidly changing channel.

For many situations, of which a cellular radiotelephone is merely an example, the prior art has not produced an apparatus or method of symbol recovery in a time-varying signal to meet the difficult requirement of providing an adaptive filter with simple and stable characteristics while maintaining fast training and recovery characteristics.

Summary of the Invention

A radio frequency receiver receives a plurality of sequential symbols transmitted from a remote signal source. The plurality of sequential symbols has a first synchronous codeword. A predetermined number of sequential symbols forms a data packet including the first synchronous codeword. The receiver stores the data packet. The stored data packet is sampled. The sampled data packet is adaptively filtered using a first filtering algorithm and a second filtering algorithm, to produce a filtered data packet. The filtered data packet is serially recovered to produce a recovered data packet.

Brief Description of the Drawings

FIG. 1 illustrates one data packet in a time division multiple access (TDMA) channel.

FIG. 2 is a block diagram for a radio telephone transceiver constructed in accordance with the present invention for receiving the data packet of FIG.1.

FIG.3 is a block diagram of the decision feedback equalizer of FIG. 2 constructed in accordance with the present invention.

FIGS. 4A and 4B describe the decision process carried out in the digital signal processor (DSP) of FIG. 2 for adaptively filtering the symbols in the data packet of FIG. 1 using multiple filtering algorithms.

Detailed Description of a Preferred Embodiment

FIG. 1 illustrates the format for a data packet 101 in a TDMA system. A receiver 201 (shown in FIG. 2) selectively receives digital information transmitted from a remote signal source 202 (shown in FIG. 2). The digital information includes a plurality of sequential symbols. A predetermined number of sequential symbols forms a data packet 101 having information for the receiver 201. A primary feature of the preferred embodiment of the present invention is that the data packet 101 is adaptively filtered using multiple filtering algorithms rather than only a conventional single filtering algorithm to remove channel-induced distortion.

Adaptively filtering the data packet 101 using multiple filtering algorithms is advantageous for receiving long data packets transmitted in a time-varying channel. In the preferred embodiment of the present invention, the RLS filtering algorithm is used over a portion of the data packet 101 for its fast training and recovery characteristics and the LMS filtering algorithm is used over a portion of the data packet 101 for its stability and low complexity characteristics. Thus, the combined filtering algorithm's complexity is reduced and the stability is improved while maintaining the fast training and recovery characteristics. In the preferred embodiment of the present invention, the bit error rate performance for symbol recovery using the RLS and the LMS filtering algorithms in combination is substantially the same as using only the RLS filtering algorithm. A receiver 201 employing the present invention may offer improved audio quality, receiver control operation or data

reception for the received data packet transmitted in a time-varying channel.

In the TDMA system, the data packet 101 includes a codeword providing synchronization, a desired slot sync word (DSSW), and a coded digital voice color code (CDVCC). A second data packet 102 adjacent to the data packet 101 also has a codeword providing synchronization designated as the adjacent slot sync word (ASSW) because of its location adjacent to the data packet 101. In accordance with the preferred embodiment of the present invention, the data packet 101 is divided into four regions (A, B, C and D) for symbol training and recovery, each region (A, B, C and D) is adjacent to one of the three codewords. The symbols in each region (A, B, C and D) are adaptively filtered using multiple filtering algorithms.

The data packet 101 includes in sequential order: the DSSW having 14 symbols, a slow associated control channel (SACCH) having 6 symbols, 65 symbols of data, the CDVCC having 6 symbols, another 65 symbols of data, and 6 symbols reserved (RSV) for future use. Sequentially following the RSV symbols is the ASSW having 14 symbols residing in the adjacent data packet 102. The DSSW and the ASSW are typically used for synchronization, equalizer retraining and time slot verification of the data packet 101 and the adjacent data packet 102, respectively, as described in the standard per §1.2.4. The DSSW and the ASSW have good autocorrelation properties to facilitate synchronization and training. The CDVCC described in the standard §1.2.5 provides the receiver 201 with channel control information.

Region A includes the fourteen DSSW symbols, six SACCH symbols and twenty nine data symbols 113 and

115. Region B includes the six CDVCC symbols and thirty six data symbols 117 and 119 on the left side of the CDVCC. Region C includes the six CDVCC symbols and thirty six data symbols 121 and 123 on the right side of the CDVCC. Region D includes the fourteen ASSW symbols, six RSV symbols and twenty nine data symbols 125 and 127.

The symbols in region A are recovered in a forward direction in time using a first sampling point determined from correlating the receiver 201 to the DSSW. The symbols in region B are recovered in a reverse direction in time using a second sampling point determined from correlating the receiver 201 to the CDVCC. The symbols in region C are recovered in a forward direction in time using the second sampling point determined from correlating the receiver 201 to the CDVCC. The symbols in region D are recovered in a reverse direction in time using a third sampling point determined from correlating the receiver to the ASSW. Data recovery in the forward and backward direction in time is described in the instant assignee's co-pending application, serial no. 9217656.9 filed on December 20, 1991, invented by Henry L. Kazecki and James C. Baker, entitled "Apparatus and Method for Recovering a Time-Varying Symbol in a Serial Data System". Data recovery using multiple sampling points is described in international patent application no. PCT/US92/06995, entitled "Apparatus and Method for Recovering a Time-Varying Signal Using Multiple Sampling Points", a copy of which has been filed at the UK Patent Office with this application, labelled "Annex 1".

In a preferred embodiment of the present invention the symbols in region A are filtered first with the RLS filtering algorithm for the fourteen DSSW

symbols, the six SACCH symbols and the eleven data symbols 113 then switching to the LMS filtering algorithm for the following eighteen data symbols 115. The symbols in region B are filtered first with the RLS filtering algorithm for the six CDVCC symbols and the eighteen data symbols 117 then switching to the LMS filtering algorithm for the following eighteen data symbols 119. The symbols in region C are filtered first with the RLS filtering algorithm for the six CDVCC symbols and the eighteen data symbols 121 then switching to the LMS filtering algorithm for the following eighteen data symbols 123. The symbols in region D are filtered first with the RLS filtering algorithm for the fourteen ASSW symbols, the six RSV symbols and the eleven data symbols 125 then switching to the LMS filtering algorithm for the following eighteen data symbols 127.

The switch from the RLS filtering algorithm to the LMS filtering algorithm is made at a predetermined point in time determined by counting the symbols in the data packet 101. The RLS filtering algorithm is used first because of its fast training and recovery characteristics. In the data packet 101 having one hundred sixty two symbols, ninety-six symbols are filtered using the RLS filtering algorithm (the six CDVCC symbols are filtered twice for training) and seventy-two symbols are filtered using the LMS filtering algorithm. Using the LMS filtering algorithm for a substantial number of symbols in the data packet 101 results in a substantial reduction in the processing time for symbol recovery and increased filter stability.

An alternate embodiment of the present invention adaptively switches from the RLS and the LMS filtering algorithm responsive to the quality of an error signal (e_k

in FIG. 3) in the adaptive filter rather than by counting the symbols. The error signal e_k is determined from the difference between a desired signal r_k and the filtered output signal y_k. The error signal e_k is affected by the quality of the received signal, i.e. the data packet 101, and the instability generated by the RLS filtering algorithm. For example, using the RLS filtering algorithm can increase the error signal level if it is used in a nondelay spread channel. Also, using the RLS filtering algorithm in a delay spread channel over multiple iterations can cause instability thereby increasing the error signal. As in the preferred embodiment a fast converging filtering algorithm such as the RLS filtering algorithm is used first to quickly train the filter 203. The error signal e_k is compared to a predetermined threshold signal in the filtering algorithm. If the error signal is below the threshold signal, the adaptive filter continues to use the RLS filtering algorithm. If the error signal is above the threshold signal, the adaptive filter switches to the LMS filtering algorithm.

The present invention is not intended to be restricted to the description of the preferred embodiment. The number and type of adaptive filtering algorithms used in accordance need not be limited to the RLS filtering algorithm and the LMS filtering algorithm. Depending on such factors as the time duration of the data packet 101, the type of channel-induced distortion, and the requirements of the receiver 201, various types and the number (two or more) of adaptive filter filtering algorithms may be used. Some other types of adaptive filtering algorithms are described in John Proakis, Digital Communications, Chapter 8 (1989).

The format of the regions (A, B, C and D) are not intended to be limited to the specified number of regions or the number of symbols per region. Rather, any number of regions, including one, may be used to filter the received information. Also, the data packet 101 is not restricted to be formatted in accordance with the IS-54 digital standard. Rather, any like signal format may embody the present invention.

A block diagram of a cellular subscriber radiotelephone 200 employing the present invention is shown in FIG. 2. Signal information including the data packet 101 is transmitted from the remote signal source 202, such as a cellular base station, to the radiotelephone 200. In an alternate embodiment, the receiver 201 may reside in a cellular base station and the remote signal source 202 would represent a cellular subscriber radiotelephone.

The data packet 101 of FIG. 1 is coupled to a duplex filter 205 via an antenna 207. The duplex filter 205 separates the received and transmit band of frequencies such that a signal may be received at the same time another signal is transmitted.

An IF receiving stage 209, coupled to the duplex filter 205, comprises a filter that is more selective than the duplex filter to generate an IF signal. The IF stage 209 separates the frequency of the data packet 101 from other radio frequency signals in the receive band presented at the antenna 207. An analog-to-digital (A/D) converter 211 converts the IF signal from the IF stage 209 in an analog format to a digital format for a data buffer 213.

The data buffer 213 stores the data packet 101 for selective processing by a digital signal processor (DSP) 255, such as DSP/556001, provided by Motorola, Inc.

The use of the DSP/56001 is described in DSP/56001 Digital Signal Processor User Manual, revision 1, available from Motorola, Inc.

A sampling point detector 215 provides an optimum sampling point for symbol recovery of the data packet 101. A controller 217 sends a slot sync signal at line 219 for the sampling point detector 215 to synchronize the receiver 201 to the data packet 101. The sampling point detector 215 then trains the receiver 201 on the DSSW in the data packet 101 responsive to a train/recover signal at line 221 set to a training mode by the controller 217. In the training mode, the sampling point detector 215 uses a reference vector r_V from the reference vector buffer 223. In the training mode, the decision feedback equalizer 203 also uses the reference vector r_V from the reference vector buffer 223 responsive to the train-recover signal 221 biasing a switch 225 to couple the reference vector r_V to the decision feedback equalizer (DFE) 203.

The sampling point detector 215 couples a symbol clock signal at line 227 to the controller 217. The controller 217 keeps track of which symbol is being processed in the sampling point detector 215 via the symbol clock signal at line 227.

After training, the controller 217 switches the train/recover signal at line 221 to a recovery state which couples a recovered signal r_S to the DFE 203 via the switch 225. The symbols in the data packet 101 are recovered in a forward and backward direction in time responsive to a forward/backward signal at line 229 coupled from the controller 217 to the sampling point detector 215. The symbols in the data packet 101 are recovered using either the RLS filtering algorithm or the LMS filtering algorithm responsive to an RLS/LMS select

signal at line 231 coupled from the controller 217 to the DFE 203. The controller 217 decides at what point in time to switch between the forward and backward recovery modes and between the RLS and the LMS filtering algorithms responsive to the symbol clock signal at line 227 from the sampling point detector 215. A symbol locator signal at line 233 generated by the controller 217 lets the sampling point detector 215 know at which symbol to begin forward or backward recovery.

The sampling point detector 215 produces sampled symbols x_k corresponding to the symbols in the data packet 101. The sampled symbols x_k are adaptively filtered by the decision feedback equalizer (DFE) 203 to produce corresponding filtered symbols y_k. The DFE 203 is a particular type of an adaptive filter. Other types of adaptive filters may be used in accordance with the present invention. A variety of adaptive equalizers are described in IEEE (Institute of Electrical and Electronic Engineers) Communication Magazine, March 1982, in an article entitled, "Adaptive Equalization" by Shahid Qureshi, pages 9 through 16.

The controller 217 signals the DFE 203 via line 231 which adaptive filtering algorithm (RLS or LMS) to use for filtering the sampled symbol x_k responsive to the symbol clock signal at line 227. The DFE tracks the phase of the data packet 101 and cancels the distortion caused by the delayed version of the data packet 101. DFE's are generally described in John Proakis, Digital Communications, Chapter 6 (1989).

The filtered symbols y_k are coupled from DFE 203 to a symbol recovery processor 235. A symbol recovery processor 235 translates the symbol to a position on the eight point constellation, representing a recovered

symbol. The symbol position on the eight point constellation represents two bits of binary information. The symbol recovery processor 235 is a conventional coherent detector used in digital communications for symbol recovery. Coherent detection is described in Bernard Sklar, Digital Communications, Fundamentals and Applications, Chapter 3 (1988).

A binary data processor 237 receives the recovered symbol from the symbol recovery processor 235 to produce corresponding formatted signals. The binary data processor 237 separates recovered voice information from recovered control information. Voice output signals are coupled to a vocoder 239 that decode the voice output signal to produce a digital representation of the voice output signal. The digital voice signal is converted to an analog voice signal in a D/A converter 241. The analog voice signal is coupled to a speaker 243, providing audible voice. The control output signals from the binary data processor 237 are coupled to a control unit 245 having a keypad and a display.

Voice and control information may also be transmitted by the cellular radiotelephone 200. Per the IS-54 standard, the content of the plurality of sequential symbols transmitted is different from content of the plurality of sequential symbols received. An audible voice message coupled to a microphone 247 produces an analog voice signal that is converted to a digital voice signal in a A/D converter 249. The digital voice signal is encoded into symbol information by the vocoder 239. The encoded symbol is formatted into the data packet 101 with any control information from the control unit 245 via the binary data processor 237. The symbols in the formatted data packet 101 are coupled to

a D/A converter 251 to produce an analog signal. The analog signal from the D/A converter 251 is transmitted by a transmitter 253 through the duplex filter 205 from the antenna 207 for reception by the remote signal source 202.

The sampling point detector 215, the controller 217, the decision feedback equalizer 203, the switch 225, and the symbol recovery processor 235 are employed in the digital signal processor 255, the data buffer 213 and the reference vector buffer 223 are memory portions of a conventional random access memory (RAM).

FIG. 3 is a block diagram of the decision feedback equalizer 203 of FIG. 2 constructed in accordance with the present invention. The DFE 203 is a simple nonlinear equalizer that is particularly useful for channels with severe amplitude distortion. The DFE 203 uses decision feedback to cancel the interference from the symbols which have already been detected. The equalized signal y_k is the sum of the outputs of feedforward and feedback parts of the equalizer. The sampled symbol x_k from the sampling point detector 215 is processed through the delay line with T-second taps (where T is the symbol duration). The outputs of the taps 301 through 304 are linearly weighted by equalizer coefficients C_0 through C_4, respectively, and summed in summer 305 to produce the equalized output signal y_k. The difference between a desired reference signal r_k and the output signal y_k is determined in a summer 306 to produce the error signal e_k. The error signal e_k is coupled to the RLS and LMS coefficient determinators 307 and 309, respectively. The tap coefficients are set to subtract the effects of interference from symbols that are adjacent in time to the desired symbol.

Decisions made on the equalized signal y_k fed back via the transversal filter 304. The basic priciple of the DFE is that if the value of the symbols already detected are known (past decisions are assumed to be correct), then the distortion contributed by these symbols can be cancelled exactly, by subtracting past symbol values with appropriate weighting from the equalizer output.

The criterion for selecting the tap coefficients C_n is typically based on the minimization of either peak distortion or mean-square distortion. Using adaptive equalization, the coefficients C_n are continually and automatically adjusted directly from the received data. The equalized symbols y_k of the DFE are expressed in terms of the sampled input symbols x_k and the tap coefficients C_n as:

$$y_k = \Sigma_{n=0}^{3} C_n x_{k-n} + C_4 r_{k-1}$$

where r_k is either r_v or r_s.

The coefficients C_n for the DFE 203 are generated by either the RLS coefficient determinator 307 or the LMS coefficient determinator 309. The determination of which coefficient determinator to use is made based on the state of the RLS/LMS select signal at line 231 from the controller 217. If the state of the RLS/LMS select signal at line 231 is high, the RLS coefficient determinator is enabled and LMS coefficient determinator is disabled. An invertor 311 causes the state of the RLS/LMS select signal to be inverted for the LMS coefficient determinator 309. Likewise, if the state of the RLS/LMS select signal at line 231 is low the RLS coefficient determinator 307 is disabled and the LMS coefficient determinator 309 is enabled. Both the RLS coefficient determinator 307 and the LMS

coefficient determinator 309 generate the coefficients C_n responsive to the sampled symbol x_k and the error signal e_k. The RLS coefficient determinator 307 also uses the desired signal r_k (r_s or the reference vector r_v depending on whether the equalizer is in a recovery or training mode, respectively). The coefficients C_n for the RLS and the LMS filtering algorithms may be determined using equations described in John Proakis, Digital Communications, Chapter 6 (1989)

FIG. 4A and 4B describe the decision flow process carried out in the DSP 255. At the block 401 in FIG. 4A, symbols are retrieved from the data packet 101 that are stored in the data buffer 213. At block 403, the sampling point is detected in the sampling point detector 215 for each symbol in the data packet. The rest of the decision flow process in FIGS. 4A and 4B may be grouped into four decision portions corresponding to the four regions A-D. Blocks 409, 411, 413, 415, and 417 are used to process region A. Blocks 419, 421, 423, 425, and 427 are used to process region B. Blocks 431, 433, 435, 437, and 439 are used to process region C. Blocks 441, 443, 445, 447, and 449 are used to process region D.

The symbols in region A of the data packet 101 are adaptively filtered forward in time. The RLS filtering algorithm is used to train the DFE 203 on the DSSW in the data packet 101 at block 409 and then continues recovering the six SACCH symbols and the eleven data symbols forward in time at block 411. The controller 217 determines when to switch between the training and the recovery mode via the train/recovery signal at line 221. At block 413 a determination is made if it is time to switch from the RLS filtering algorithm to the LMS filtering algorithm. If the determination is

negative the flow returns to block 411 whereby the RLS filtering algorithm is used to continue recovering the symbols. If the determination is positive, the eighteen data symbols 115 in the data packet 101 of FIG. 1 are recovered forward in time using the LMS filtering algorithm. The controller 217 determines when to switch to the LMS filtering algorithm at block 413 responsive to the symbol clock signal at line 227 of FIG. 2. The decision to switch is made by the controller 217 by counting the number of symbol clock signals or by comparing the error signal to the threshold signal. The controller 217 enables the LMS filtering algorithm and disables the RLS filtering algorithm via the RLS/LMS select signal at line 231. A determination is made at block 417 if it is time to train the DFE 203 on the CDVCC backward in time. If the determination is negative, the flow returns to block 415 to continue recovering the data symbols 115 using the LMS filtering algorithm. If the determination is positive, the flow continues to block 419.

The symbols in region B are adaptively filtered backward in time. The decision process continues with the CVDCC in the data packet responsive to the symbol locator signal at line 233 generated by the controller 217 in FIG. 2. Using the RLS filtering algorithm, the DFE is trained on the CVDCC at block 419 and continues recovering data symbols 117 backward in time at block 421. A determination is then made at block 423 if it is time to switch to the LMS filtering algorithm. If the determination is negative, the flow returns to block 421 whereby the data symbols 117 are recovered backward in time. If the determination is positive at block 423, the DFE 203 switches from using the RLS filtering algorithm to the LMS filtering algorithm responsive to

the RLS/LMS select signal at line 231 to recover the data symbols 119 backward in time at block 425. At block 427 a determination is made if it is time to train the DFE 203 on the CDVCC forward in time. If the determination at block 427 is negative, the flow continues to block 425 wherein the data symbols 119 are recovered using the LMS filtering algorithm backward in time. If the determination at block 427 is positive, the flow continues to block 431 in FIG. 4B.

The symbols in region C are adaptively filtered forward in time. Using the RLS filtering algorithm, the DFE is trained on the CDVCC at block 431 and the data symbols 121 are recovered forward in time at block 433. At block 435, a determination is made if it is time to switch to the LMS filtering algorithm. If the determination is negative at block 435, the flow returns to block 433 whereby the data symbols 121 are recovered forward in time. If the determination at block 435 is positive, the flow continues to block 437 wherein the data symbols 123 are recovered forward in time using the LMS filtering algorithm. At block 439 a determination is made if it is time to train the DFE 203 on the ASSW backward in time. If the determination is negative at block 439, the flow returns to block 437 wherein the data symbols continue to be recovered using the LMS filtering algorithm forward in time. If the determination at block 439 is positive, the flow continues to block 441.

The symbols in region D are adaptively filtered backward in time. Using the RLS filtering algorithm, the DFE 203 is trained on the ASSW at block 441 and continues recovering the six RSV symbols and the data symbols 125 backward in time. At block 445, a determination is made if it is time to switch from the

RLS filtering algorithm to the LMS filtering algorithm. If the determination at block 445 is negative, the flow returns to block 443 wherein the symbols continue to be recovered using the RLS filtering algorithm. If the determination at block 445 is positive, the flow continues to block 447 whereby the data symbols 127 are recovered backward in time using the LMS filtering algorithm. While each of the 18 data symbols 127 are recovered, a determination is made at block 449 if the last symbol in the data packet has been recovered. If the determination is negative at block 449, the flow returns to block 447 whereby the data symbols 127 continue to be recovered using the LMS filtering algorithm backward in time. If the determination at block 449 is positive, the flow continues to block 451 wherein other receiver functions are processed such as mobile assisted handoff (MAHO) and voice or data determination.

Thus, a receiver 201 having an adaptive filter uses multiple filtering algorithms to train and recover information in a time-varying channel. The switch from one filtering algorithm to the other may be responsive to a predetermined location of a symbol in the data packet or the quality of the error signal in the adaptive filter. In the preferred embodiment of the present invention, multiple filtering algorithms were used to minimize processing time in the DSP while providing substantially equivalent bit error rate performance. In other applications where processing time is not as critical, other advantages may be gained such as improving the bit error rate performance and improving the stability of the adaptive filter.

What is claimed is:

Claims

1. Adaptive filter means for removing distortion in an input signal, having a plurality of sequential symbols, responsive to at least the input signal, an output signal, a reference signal and a plurality of coefficients, the adaptive filtering means comprising:

means for determining a first set of coefficients using a first filtering algorithm; and

means for determining a second set of coefficients using a second filtering algorithm.

2. The adaptive filter means in accordance with claim 1 further comprising:

means for detecting the predetermined position of at least one symbol; and

means for selecting, responsive to the means for detecting, between the first filtering algorithm and the second filtering algorithm.

3. The adaptive filter means in accordance with claim 1 further comprising:

means for determining the difference between the reference signal and the output signal to produce an error signal;

means for comparing the error signal to a threshold signal; and

means for selecting, responsive to the means for comparing, between the first filtering algorithm and the second filtering algorithm.

4. The adaptive filter means in accordance with claim 1 further comprising means for dividing the input signal into at least a first and a second region, a first portion of each region is adaptively filtered using the first filtering algorithm, a second portion of each region adaptively filtered using the second filtering algorithm.

5. A method for removing distortion in an input signal, having a plurality of sequential symbols, responsive to at least the input signal, an output signal, a reference signal and a plurality of coefficients, the method comprising the steps of:

determining a first set of coefficients using a first filtering algorithm; and

determining a second set of coefficients using a second filtering algorithm.

6. A radio frequency receiver for receiving a plurality of sequential symbols transmitted from a remote signal source, the plurality of sequential symbols having at least a first synchronous codeword, a predetermined number of sequential symbols forming a data packet including at least the first synchronous codeword, the radio frequency receiver comprising:

means for storing the data packet;

means for sampling the stored data packet;

means for adaptively filtering the sampled data packet using a first filtering algorithm and a second filtering algorithm, to produce a filtered data packet; and

means for serially recovering the filtered data packet to produce a recovered data packet.

7. The radio frequency receiver in accordance with claim 6 further comprising:

means for detecting the predetermined position of at least one symbol in the data packet; and

means for selecting, responsive to the means for detecting, between the first filtering algorithm and the second filtering algorithm.

8. The radio frequency receiver in accordance with claim 6 further comprising:

means for determining the difference between a desired symbol and a filtered symbol in the filtered data packet to produce an error signal;

means for comparing the error signal to a threshold signal; and

means for selecting, responsive to the means for comparing, between the first filtering algorithm and the second filtering algorithm.

9. The radio frequency receiver in accordance with claim 6 further comprising means for dividing the data packet into at least a first and a second region, a first portion of each region is adaptively filtered using the first filtering algorithm, a second portion of each region adaptively filtered using the second filtering algorithm.

10. A method for receiving a plurality of sequential symbols transmitted from a remote signal source, the plurality of sequential symbols having at least a first synchronous codeword, a predetermined number of sequential symbols forming a data packet including at least the first synchronous codeword, the method comprising the steps of:

storing the data packet;

sampling the stored data packet;

adaptively filtering the sampled data packet using a first filtering algorithm and a second filtering algorithm, to produce a filtered data packet; and

serially recovering the filtered data packet to produce a recovered data packet.

Patents Act 1977
Examiner's report to the Comptroller under Section 17 (The Search Report)

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K WILLIAMS

Databases (see over)

(i) UK Patent Office

(ii) ONLINE DATABASE: WPI

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Documents considered relevant following a search in respect of claims 1-10

Category (see over)	Identity of document and relevant passages	Relevant to claim(s)
X	GB 2236036 A (NOKIA DATA SYSTEMS) see page 8, lines 11-32	1,5,6,10
X	EP 0426026 A2 (MITSUBISHI DENKI) see Claim 1	1-10
X	EP 0345675 A2 (NATIONAL SEMICONDUCTOR CORP) see Claims 1 and 2	1-10
X	EP 0203726 A1 (BRITISH TELECOM) see page 9, lines 7-11	1-10
X	US 4633482 (U S PHILIPS CORP) see Claim 1 & EP 0146979	1-10

Category	Identity of document and relevant passages	Relevant to claim(s.

Categories of documents

X: Document indicating lack of novelty or of inventive step.

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P: Document published on or after the declared priority date but before the filing date of the present application.

E: Patent document published on or after, but with priority date earlier than, the filing date of the present application.

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